Two-Channel Stereophonic Sound Systems

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Basic Requirements for Realistic Sound Location

THE aim of a perfect stereophonic sound-reproducing system is to create for the listeners a similar sound picture in correct aural perspective to that which they would have if transported to an ideal position from which to hear the original sounds. Although this might be possible using a multi-channel system, two channels can, at the best, only recreate the original sounds in correct aural perspective over a limited distance bounded by the two loudspeakers. This is, however, a big improvement over the reproduction available from a single channel system and results in a considerable increase in realism and clarity of the reproduction sound.

The article considers first the particular information used by the brain for the location of sounds, and from a study of this information a two-channel system is devised. In particular a simple method is given for correcting for various listening positions, and it is also shown that the reproduced sound image is more accurately positioned if arrival time differences are overruled and the sound is positioned by intensity differences only.

Information Available for Sound Location.—This subject is dealt with fully in the literature (see, for example, refs. 1, 2, 3). Summarizing the findings of the numerous measurements made it can be said that the brain appears to make use of the following factors:—

(i) Relative loudness of the sound at the two ears.
(ii) Differences in the sound spectrum in the two ears.
(iii) Relative time of arrival and relative phase of sound at the two ears.
(iv) The "quality" of the source as compared with previous knowledge of the quality of a similar source.
(v) The differences both in quality and time of arrival of the direct sound with any reflected sound.

For left-right perception factors (i), (ii), and (iii) appear to be the most important whilst front-back perception and distance perception rely mainly on factors (iv) and (v).

The brain can make use of all the information supplied to it by the ears and the best sound location occurs when all the information is in the same sense. As an example of information in a contradictory sense, in a very live room the reverberation may be louder than the direct sound and arrive from a different position, but by taking into account the late time of arrival of the reverberation the brain is able to ignore the reverberation and to ascertain the true position of the source of sound. This position will not be quite as well defined as it would have been with no reverberation, but it will still be fairly accurate.

Before describing any stereophonic systems, one point of possible confusion should be settled. Throughout these descriptions the term "time difference" between two sounds will be used, and no use of the term "phase difference" will be made. This is because the first is meaningful to random and transient sounds and ambiguous for repetitive waves, whilst the second is relatively meaningless for random and transient sounds but applicable to repetitive waves. Since directional location and stereophonic effects are very much better for random and transient sounds the time difference concept will be used.

Experimental Two-channel system.—Fig. (1) shows the layout of an experimental two-channel system set-up to investigate various effects. It is necessary to give the system an exact size because whilst intensity differences are relative, time delay is a scalar magnitude and will not alter in the same way as intensity differences if the experimental system is made larger or smaller. It was decided at the outset that an attempt would be made to obtain stereophonic reproduction over a front of ten feet and that the listener should be situated somewhere on a line ten feet long and eight feet or more away from the base line of the two loudspeakers.

The first experiment was to ascertain the extent to which two loudspeakers, placed ten feet apart, could simulate a single source of sound when heard by a listener in position L3, that is, equidistant from both loudspeakers and facing the centre of the loudspeaker base line. It was found that with the sound levels from the two loudspeakers equal the listener perceived an apparent single sound source straight in front of him. Extreme movement of the head by the listener tended to produce a splitting of the sound image into two separate sources, but fortunately this effect is not serious unless the listener is consciously trying to listen to each of the loudspeakers independently.

Now if the observer moves from listening position L3 to position L2, then the virtual sound image moves towards one side, in this case towards LSA. This is as would be expected since LSA is now nearer the listener than LSB, and hence the intensity of the sound from LSA is greater and has an arrival time in advance of that from LSB. This is the most serious and the most often encountered fault in two-channel stereophonic systems. It means that the system works correctly only for listening positions on the centre line between the two loudspeakers. However, within limits this fault can be compensated for by the use of directional loudspeakers.

Considering listening position L1 with both loud-

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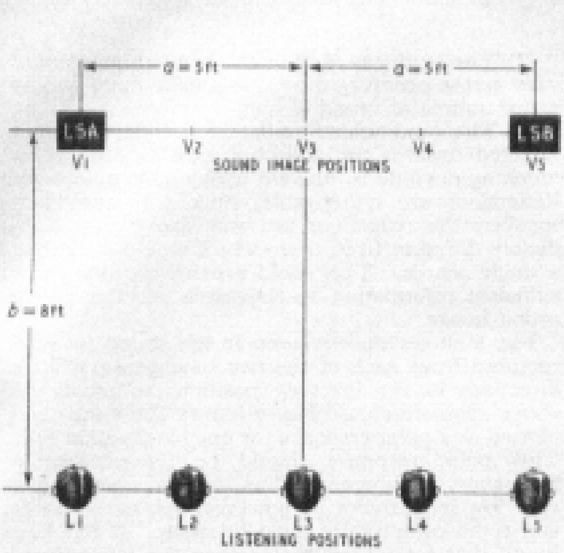


Fig. 1. Layout of experimental stereophonic system in an echo-free room. In this test the loudspeakers LSA and LSB are non-directional in the horizontal plane.

speakers radiating the same random sound at the same level then

Distance LSA to L1 $= b$

Distance LSB to L1 $= \sqrt{b^2 + 4a^2}$

$\therefore$ At listening position L1 the sound intensity due to LSA will be greater than that due to LSB by the ratio

$$20 \log_{10}\sqrt{\frac{b^2 + 4a^2}{b}}\ \text{dB}$$

$$= 10 \log_{10}\left(1 + \frac{4a^2}{b^2}\right)\ \text{dB}$$

Also the sound from LSA will be in time advance compared with the sound from LSB by an amount proportional to $\sqrt{b^2 + 4a^2} - b$

If the distances are measured in feet, the numerical value gives the approximate time advance in milliseconds.

Correction for Position of Observer.—It has been found that it is possible to correct for both these time and intensity differences and to restore the virtual sound image to the mid position by increasing the sound from LSB and decreasing it from LSA. This implies that time of arrival differences can be compensated by sound intensity differences. Such correction has been found possible for time differences up to a maximum of about five milliseconds, after which the position of the virtual sound image becomes less well defined and compensation for time differences greater than ten milliseconds becomes impossible since the sound then splits up into two distinct sources.

By the above method it is therefore possible to obtain good stereophonic reproduction for a line parallel to the speaker base line as well as for the central listening positions. However, using this method of compensation it is found that the stereophonic reproduction is also greatly improved for the area behind the corrected line and hence it is possible to cover an area with satisfactory stereophonic sound.

Experimental Results.—The test procedure was as follows: the listener was first seated in position L3 and asked to name the position from which the sound appeared to come for different relative sound levels from the two loudspeakers. The sound consisted of a short portion of speech of a few seconds duration only. Fig. 2(a) shows the results of these experiments, each point being the average for a number of listeners, each listener making several determinations for a variety of conditions of differences between the loudspeaker sound levels.

The test was then repeated with the listeners in turn in position L1, the results being shown in Fig. 2(b). Note that in this case of "offset" listening the sound intensity at loudspeaker LSB was always greater than that at LSA but that the overall shape of the curve is the same as that for the central listening position but displaced. As a result of these and other tests for the other listening positions it

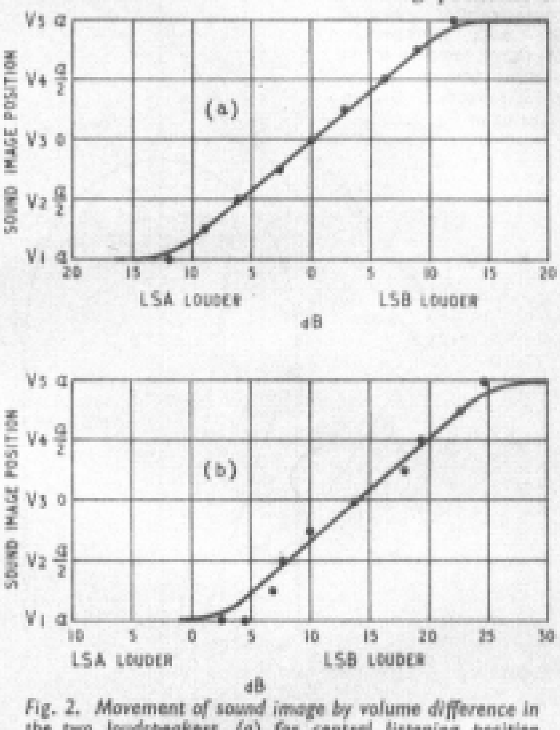


Fig. 2. Movement of sound image by volume difference in the two loudspeakers, (a) for central listening position L3 and (b) for position L1.

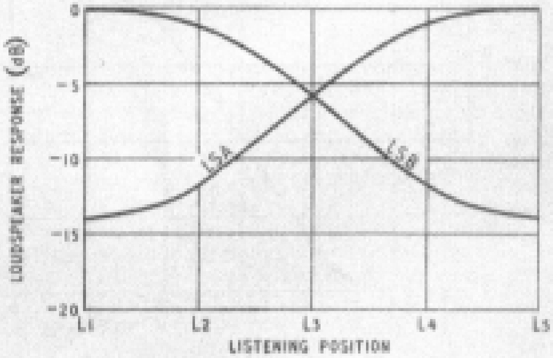


Fig. 3. Loudspeaker responses necessary to correct for off-centre listening positions (Listening line 8ft distant from loudspeakers spaced 10ft apart, as in Fig. 1).

was shown that good definition was possible along the complete listening line.

Directional Loudspeakers.—The above experimental results enable the required ratio of the sound intensities from the loudspeakers towards each listening position to be determined, so as to correct for off-centre listening. As has been indicated, this changing ratio as the listener moves from L1 to L5 can be produced by the use of directional loudspeakers. Considering only the central position for the virtual sound image, which is justified since the sound image position intensity ratio curves were all similar in shape, the requirements to be met can be summarized as:—

(1) The sound image must remain at V3 for listening positions L1 to L5.

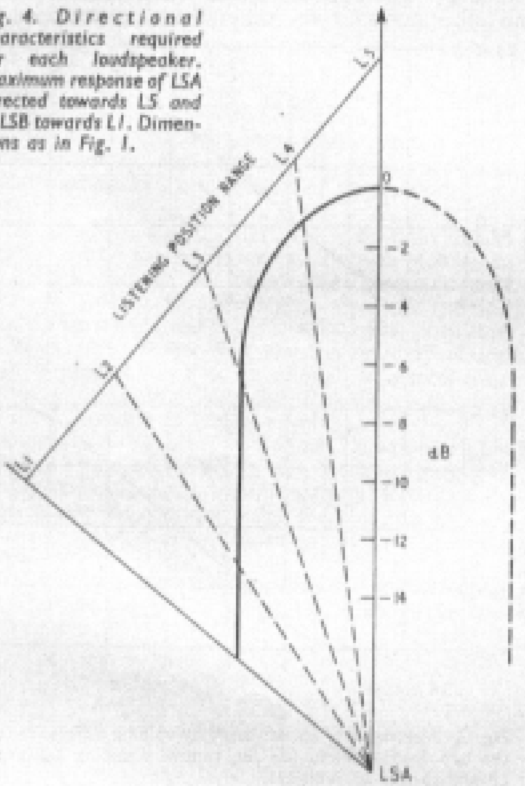
(2) The loudness of the virtual sound image must vary as the observer moves, just as if there was an actual source of sound at V3.

(3) The loudspeaker radiation outside the range directed towards the audience must be reduced as much as possible to prevent undesirable reflections. Reflections are undesirable, since with two loudspeakers the reflections are very liable to be completely different from those which would arise from a single source. This could provide the brain with sufficient information to suggest a splitting of the sound image.

Fig. 3 shows the variation in the sound intensity required from each of the two loudspeakers in the directions of the listening positions to satisfy the above requirements. Fig. 4 shows the same thing plotted as a polar response for one loudspeaker only. This polar response should be independent of frequency; a directional loudspeaker operating at very low frequencies is, however, excessively large, and some compromise must be made. It has been found that for off-centre listening the stereophonic effect does not deteriorate badly if the directionality of the loudspeakers ceases below about 300 c/s and a lower limit of even 1 kc/s provides very acceptable results.

Movement of the Virtual Sound Image.—If the sound is of the character of random noise the virtual sound image can be moved about by two methods. First, as already shown in Fig. 2(a) for an observer at the listening position L3, if the sound intensity levels of the two loudspeakers are different, then the sound image moves towards the louder source. Secondly, if the signal is retarded in time to one loudspeaker, the sound image moves towards the other loudspeaker as illustrated in Fig. 5. In addition a method employing a combination of intensity and time difference could be used and is in fact frequently encountered. It has been found, however, that a sound image moved by an intensity difference between the sound outputs from the loudspeakers remains far sharper and better defined than one where time delay is employed. From Fig. 5 it will

Fig. 4. Directional characteristics required for each loudspeaker. Maximum response of LSA directed towards L5 and of LSB towards L1. Dimensions as in Fig. 1.



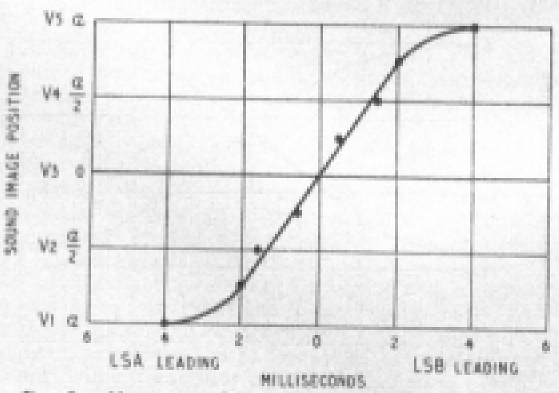
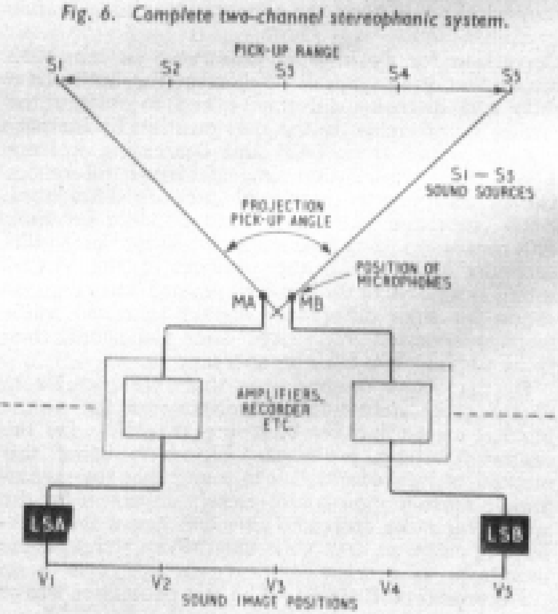


Fig. 5. Movement of the sound image at the central listening position L3 by the introduction of a time difference between the outputs from the two loudspeakers.

Fig. 6. Complete two-channel stereophonic system.



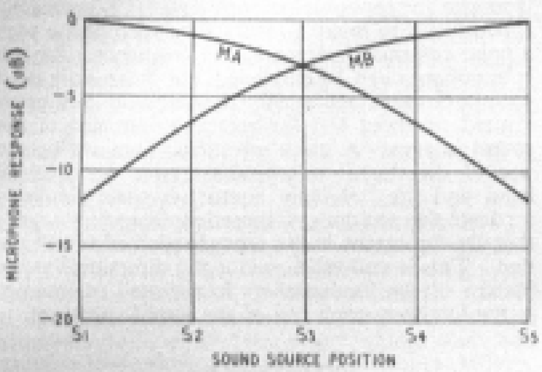


Fig. 7. Microphone responses in direction of indicated sound source for correct positioning of sound images.

be seen that for the experimental system a time delay of about four milliseconds was necessary to move the sound image over to one loudspeaker. Now consider the case if the sound had been repetitive, say with a frequency of 500 c/s. With the loudspeakers radiating the same signal both in intensity and time, the virtual sound image would have been half-way between the loudspeakers. On applying a time delay to one loudspeaker the image would have moved towards the other loudspeaker; however, had the delay been increased to two milliseconds (i.e., one period of the waveform) conditions would have returned to the state of no delay and the sound image would have returned to the central position. Now if the sound had been random or transient in nature it would have remained displaced. Hence for a sound like a piano note which has both a transient part and a fairly steady repetitive component considerable ambiguity would exist as to the exact location of the sound. In practice this effect manifests itself as an apparent widening of the sound image and also as an apparent movement of musical instruments as different notes are played. Hence moving the position of the virtual sound image by varying the intensity levels of the sounds from the loudspeakers is to be preferred.

Sound Pick-up—Microphone Polar Response.— Since it has been shown that the inputs to the two loudspeakers should have a difference of level only and not of arrival time, it follows that the two pick-up microphones should be placed close together to avoid time differences. This in turn calls for some form of directional characteristic or "shadowing" in order that the microphones may differentiate between sound arriving from the left or the right. Before investigating the methods by which these directional characteristics can be produced, it is necessary to ascertain the exact directional characteristics required. Referring to Fig. 6, it is necessary to line up the virtual sound images positions V1 to V5 at the listening end with the actual sound source positions S1 to S5 at the pick-up end of the system. The ratio between the sound intensities from the loudspeakers to position correctly the virtual sound images can be found from Fig. 2(a) and hence the necessary ratio of the responses of the microphones towards each actual sound source can be calculated. At the same time the total output from the loudspeakers must be such that as the virtual sound image is moved from V1 to V5 the loudness changes as if an actual source was moved.

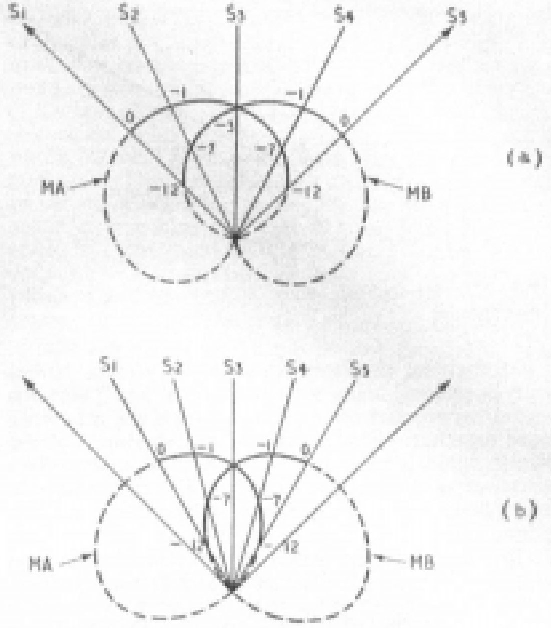


Fig. 8. Polar response curves for microphones for projected pick-up angle (a) of 90° and (b) of 60°.

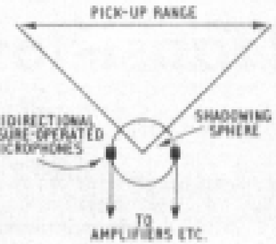


Fig. 9. Stereophonic sound pick-up using artificial head.

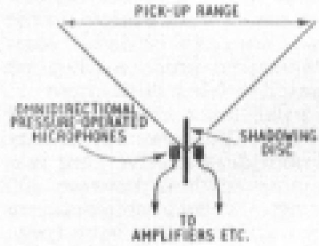


Fig. 10. Stereophonic sound pick-up using shadowing disc.

Fig. 7 shows the necessary microphone responses which satisfy these requirements. Working from Fig. 7 the necessary polar response diagrams can be drawn. Two particular examples are shown in Fig. 8, the first being for a pickup angle of 90° and the second for a pickup angle of 60°.

One of the methods of obtaining the necessary response[4, 5] is to mount two omni-directional microphones in place of the ears in an artificial head as shown in Fig. 9. Such a system depends mainly on intensity difference operated at high frequencies only. Time differences play little part since the maximum time difference which can be obtained from a head of average size is only about 0.6 milliseconds and, as can be seen from Fig. 5, this amount produces little movement.

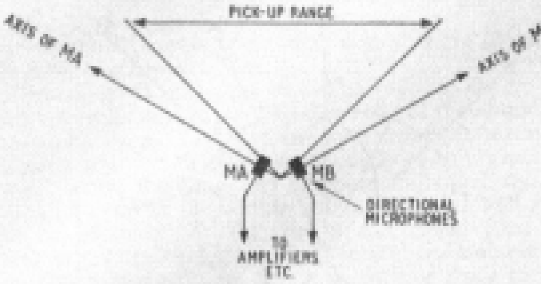


Fig. 11. Stereophonic sound pick-up using directional microphones.

A somewhat similar method shown in Fig. 10 has been developed using a flat "shadow disc" between pressure operated microphones which are otherwise close together. The effect of the "shadow" of the disc is again largely influenced by frequency. The system tends to produce excessive differences at high frequencies and inadequate differences at low frequencies.

Directional microphones placed close together as shown in Fig. 11 give the most satisfactory pick-up from the stereophonic point of view.* Unfortunately it is difficult to make a satisfactory microphone with a polar response independent of frequency. Ribbon microphones can be employed, but if these are used with their axes set at 90° the angle of pick-up is limited to about 60° for correct positioning of the sound images. A great advantage obtained by the use of directional microphones over the artificial head and the "shadow board" is their ability to separate the positions of those low frequency sounds that are important in the reproduction of reverberation. This is still valid even if the directional sound pattern of the loudspeakers is not well maintained at the low-frequency end of the sound spectrum.

* G.E.C. system and E.M.I. "Stereosonic" System.

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Transistor Digital Computers

NEW BINARY CIRCUIT TECHNIQUES DESCRIBED AT I.E.E. CONVENTION

WHEN the thermionic valve was introduced into digital computing it made possible machines of remarkable versatility and tremendous speed of operation but also brought with it a number of disadvantages. These were perhaps not obvious in the early days, but now, with hundreds of digital computers being sold as commercial products, they are beginning to make themselves felt a little more.

To begin with, the valve has a certain rate of failure and limitation of life. This may not be very important in a domestic broadcast receiver, but in a digital computer, containing anything between 300 and 3,000 valves, it becomes of considerable nuisance value (a graph in another article in this issue (page 232) gives some idea of how reliability of equipment decreases with number of components). Secondly, when several hundred (or thousand) valves are massed together in a single equipment they generate a great deal of heat, and so threaten the reliability of other components—not to mention the kilowatts of electric power consumed in the process. Thirdly, there is the uneconomical size of valves for digital computing operations; considering that most of them do little more than act as simple two-state elements they take up an unnecessary amount of space.

It is only to be expected, therefore, that alternative devices are being sought that will overcome these particular disadvantages. At the moment there are two principal ones—the transistor and the two-state magnetic core. Both are small and robust, produce very little heat, are efficient in operation and appear to have a long expectation of life (so far as we can tell at present). In addition they will both operate from a single source of power of only a few volts.

The possibilities of these devices, and methods of using them in digital computers, were recently discussed at a highly successful convention on digital computer techniques held by the I.E.E. in London. A whole session, in fact, was given over to "The Transistor." This included papers on two complete transistor digital computers, one built at Manchester University and the other at the Atomic Energy Research Establishment, Harwell, while later on there were papers on special computing circuits using combinations of transistors and magnetic cores.

It was interesting to note that both of the complete machines relied principally on point-contact transistors for the computing circuits, and it seems that these devices are still regarded very highly by the computer people, even though everybody else has virtually dismissed them as obsolete. The properties of the point transistor were, of course, recognized very early on as being suitable for pulse and switching circuits. In the first place there was a good frequency response, and secondly, unlike the junction transistor, a negative resistance characteristic that could be used to give a regenerative change-over action in a two-state circuit (equivalent to the Eccles-Jordan valve trigger commonly used in binary computing).*

Unfortunately the point-contact transistor proved

* See "Transistors—Applications in Trigger Circuits," by Thomas Roddam, *Wireless World*, June, 1953.